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Voice Services Options for the Battlespace Communications System (Land)

W.D. Blair and R. Jana

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ABSTRACT

The current generation Battlespace Communications System (Land) for voice (telephony) is based on a military standard circuit switching technology. There has been considerable discussion on alternatives based on packetised voice, ie voice over Internet Protocol and voice over ATM. This paper examines these two options and considers the bandwidth demands of each option, specifically in the context of an agreed reference model of the network including traffic load. This report is preliminary work prior to the combination of this voice model with that of an agreed data workload and examination through simulation of consequent integrated network performance.

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Voice Services Options for the Battlespace Communications System (Land)

Executive Summary

The Australian Defence Force has been procuring under Project Parakeet a digital communications network for use by forces deployed to the land battlefield. It employs a military standard circuit switching technology and is effectively a militarised narrowband ISDN system. The current phase of Parakeet has recognised the limitations of the circuit switching technology in carrying computer data communications. Accordingly, a concept demonstrator element has been added to the current phase that will field a limited number of Asynchronous Transfer Mode (ATM) switches along with an Internet Protocol (IP) capability which will explore the carriage of multimedia services over Parakeet. As a parallel activity, effort is being expended to make a comparative performance study between the two technologies, ATM and IP, for delivering multimedia traffic (voice, data and possibly video in the future).

This paper examines the two options of Voice over IP (VoIP) and Voice Telephony over ATM (VTOA) and considers the bandwidth demands of each option, specifically in the context of an agreed reference model of the network including voice traffic load. The study found that a simplistic VoIP approach would be extremely wasteful of the limited bandwidth. Any use of IP as a total WAN solution for voice services must implement protocol header compression to be feasible in the tactical domain. The average bandwidth demands used by the voice system during the busy hour on the busiest trunk link indicate that VTOA and VoIP with packet header compression are certainly promising solutions and more than twice as efficient as standard VoIP. Regardless of which approach is adopted, significant capacity can be released for non-real time communications.

This report is intended to be preliminary work prior to the combination of this voice model with that of an agreed data workload and examination through simulation of consequent integrated network performance. The findings of this report justify the continuation of the analysis to the next stage. The final outcome to Defence will provide significant input into the feasibility of including civil technologies into the technical architecture of a future Battlespace Communications System (Land).

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1. Introduction

The Australian Defence Force has been procuring under Project Parakeet a digital communications network for use by forces deployed to the land battlefield. The system uses various transmission systems to provide voice and data services between headquarters. It employs a military standard circuit switching technology and is effectively a militarised narrowband ISDN system. In circuit switching technology necessary resources such as bearer channels are allocated by the network for the duration of the phone call. By contrast, more modern data communications systems transport data in segments (packets or cells) which are routed around the network consuming resources in a more flexible and responsive manner.

The current phase of Parakeet has recognised the limitations of the circuit switching technology in carrying computer data communications. Accordingly, a concept demonstrator element has been added to the current phase that will field a limited number of Asynchronous Transfer Mode (ATM) switches along with an Internet Protocol (IP) capability. This concept demonstrator will be trialed and assessed in the context of developing the next generation of battlespace (land) communications systems. As a parallel activity, effort is being expended to make a comparative performance study between the two technologies, ATM and IP for delivering multimedia traffic (voice, data and possibly video in the future).

This report will first highlight its scope in relation to current studies into the Battlespace (Land) Communications Architecture. We will then review some background material on the multimedia communications standard H.323 for IP based networks specifically in the context of Voice over IP (VoIP). A similar review will be carried out on the AAL-2 standard for Voice Telephony over ATM (VTOA). Section 4 will calculate the bandwidth characteristics of VoIP and VTOA examining the network demands as a function of the number of voice calls being carried on a trunk. Section 5 will discuss signalling aspects of VoIP and VTOA. The endorsed Parakeet reference network will be introduced in Section 6 that will form the basis for determining the voice traffic load. Section 7 will combine the results of the traffic analysis and the bandwidth characterisation to deduce the peak and average network loads from the alternative approaches.

2. The Land Battlespace Communication Studies

There are a number of studies currently being conducted in the area of battlespace (land) communications. An overarching architectural study [1] is being conducted by collaboration between University College (University of NSW at the Australian Defence Force Academy) and DSTO. The aim of this work is to develop an operational concept description of the Battlespace (Land) Communications System to help shape the development of a Project Definition Study (PDS) under Project Parakeet and its follow on project under Joint Project 2072 (JP-2072).

The PDS will receive input from trials of the ATM and IP concept demonstrator being developed as part of phase 6 Project Parakeet. As a separate initiative, an

unsolicited proposal from the communications consultant company Codarra for an examination of a totally IP based land battlespace communications system led DSTO to undertake work with a prime focus on quantitative assessment of the IP WAN option.

This report will provide input to the study in respect of the voice network load. Once an agreed data traffic model is endorsed, this will be combined with the voice model to examine through simulation the consequent integrated network performance network. Performance parameters such as congestion, latency and latency variation will be particularly examined in the context of real time services such as voice.

3. VoIP and VTOA Standards

3.1 Overview

IP is the fundamental underlying protocol used on the world wide Internet. The internet protocol suite (usually known as TCP/IP from Transmission Control Protocol/Internet Protocol) has been developed using a layered approach [2]. The IP layer provides a best effort service to get data packets from the originating host to the destination host. This provides the internetworking layer upon which the well-known Transport Control Protocol (TCP) layer provides a reliable end-to-end connection service. Less well known is the User Datagram protocol (UDP). This protocol operates at the same layer as TCP, however provides a best effort datagram service over IP, better suited for the exchange of real-time information which would rather have data discarded than suffer excessive time delay. Below the IP layer is the physical layer, which in our case is likely to be the Point to Point Protocol (PPP).

ATM is quickly becoming a popular mechanism to provide a single network capable of supporting a wide range of services with a particular strength in its ability to support a guaranteed quality of service (QoS). The fundamental concept of ATM revolves around the transfer of information in small, fixed length packets called cells. Since ATM uses a connection-oriented service, the cell headers can be considerably smaller than IP packets and avoid the header overhead from becoming too burdensome. It has many advantages over circuit switching the most important of which is, like IP networks, the ability to conduct dynamic bandwidth allocation. Several data sources can be multiplexed on the same connection in a flexible fashion to allow users to obtain *bandwidth on demand.* To provide the link between user applications and the ATM cell layer, a number of alternative protocols forming the ATM Adaptation layer (AAL) have been developed. In this report, we will consider real-time voice traffic over ATM using the AAL-2 standard.

There are two fundamental approaches to implementing a voice service over a digital WAN, regardless of whether an IP or ATM mechanism is being employed. The two approaches are not mutually exclusive and can co-exist and interconnect through the use of gateways.

Individual User Interconnection. For this option each connection is treated totally independently and the audio bit stream is passed from user to user over the underlying data switching infrastructure (IP routers or ATM switches). Note it is important to

discriminate between the switching control function of a voice switch/PABX and the actual switching process. There may be a voice network controlling service on the network to route calls, determine availability of end terminals and provide value added service such as voice mail/follow me etc. However, the actual connections do not pass through the physical controller device.

PABX Interconnection. Here calls from subscribers destined for the WAN are concentrated into one point in a local area, the PABX. The interconnection of PABXs over the WAN can then be through any number of means, for the purpose of this report via VoIP or VTOA. PABX interconnection via VoIP is typically via a gateway so, from the perspective of the WAN, the connections appear to be individual user interconnections. Interconnection of PABXs via ATM is significantly different and can occur in four approaches:

- Each PABX interconnection has a separate physical port on the PABX. The port is then connected to an ATM switch and passes over a logical connection that is transparently carried over the ATM network via a constant bit rate Circuit Emulation Service (AAL-1). The other end of the interconnection terminates at a fixed port on a specific PABX. Thus the ATM network is merely emulating point-to-point links between PABXs. All the intelligence is vested in the PABX that will typically need to tandem calls to route connections through the network.
- A minor variation on the previous approach could see a single physical connection between the PABX and the ATM switch carrying a TDM group. Predetermined channels on the group are connected over the ATM to predetermined PABXs. Functionally this is the same as the first option, except that the interconnections between PABXs will have a fixed, limited number of channels, and all the interconnections emanating from a particular PABX will share one physical TDM group until the first ATM switch.
- The next stage in sophistication sees the ATM switch monitor the signalling on the PABX trunk sufficiently only to determine which TDM channels are active, pass active channels only over the ATM network, and recreate the complete TDM trunk at the distant PABX. When using AAL-1 this is known as the Dynamic Bandwidth Circuit Emulation Service. The same approach can be adopted using AAL-2 (considered to be the "non-switched trunking mode").
- The previous two approaches relied on the PABX to determine the routing of calls. A more sophisticated approach, typically using AAL-2, is to have more capable signalling processing in the interface to the ATM switch – the interworking facility (IWF). In effect the ATM network becomes a single virtual PABX. The entry ATM switch interprets the signalling coming from the PABX, determines the exit ATM switch and passes the voice channel stream with all the other calls for that exit ATM switch onto the ATM circuit. There may be no direct connection between entry and exit ATM switches, but the composite stream remains in ATM cells until exiting the ATM network. This is called the "switched trunking mode".

For the purposes of this study, the ATM option is assumed to be operating in a switched trunk mode (sophisticated PABX interconnection model). As noted earlier,

from the point of view of WAN utilisation, there is no difference between the two interconnection methods when VoIP is being used.

3.2 VoIP

3.2.1 H.323 Recommendations

H.323 is an umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over Local Area Networks (LAN) that do not provide a guaranteed QoS [3][4]. The standard is broad in scope and includes both stand-alone devices and embedded personal computer technology conducting either point-to-point connections or multipoint conferences. The H.323 standard provides a foundation for audio, video and data communications across IP based networks. By complying with H.323, multimedia products and applications from multiple vendors can interoperate, allowing users to communicate without concern for compatibility. Fundamental to H.323 is the Real-Time Protocol/Control Protocol (RTP/RTCP) of the Internet Engineering Task Force (IETF) [5] used for carrying packetised real-time traffic over an IP network. Figure 1 shows the RTP packet structure. Typically an RTP packet has a payload of 10 to 160 bytes for applications that use compressed analog to digital coding. (Note that data oriented communications and computer programmers typically use the term byte for an 8 bit data parcel. By contrast the telecommunications term for 8 bit parcel is octet. This report will use both terms interchangeably.) Analog to digital coding in a low bit rate voice environment is normally a discontinuous process. It entails taking a short sample (typically tens of milliseconds) of digitised voice and mathematically processing this waveform data into a frame of coded voice data. For a coder operating at a typical 8 kbps and sample times of 10 ms the frame would be 80 bits. There is an option within RTP to carry multiple channel frames in the payload to improve efficiency. An approach of transporting four frames in one payload is a typical commercial option but comes at the price of additional channel latency (i.e. the number of voice frames times the frame sample period).

	IP	UDP	RTP	Payload
				Channel data
bytes	**20**	**8**	**12**	**10 to 160**

Figure 1 – Real Time Services over IP

An architectural overview of the H.323 system and its components is shown in Figure 2. H.225 describes the media (audio and video) stream packetization, media stream synchronisation, control stream packetization and control message formats [6]. H.245 describes the procedures and messages used for opening and closing logical channels for audio, video and data, capability exchange, mode requests, control and

indications [7]. These are the recommendations that govern the operation of H.323 equipment and the communications between H.323 endpoints. For audio coding, G.711 is mandatory while G.722 and G.729 are optional. In this report we will evaluate the performance of voice traffic over IP (and ATM) using the ITU-T recommendation G.729 Annex B toll-quality speech coding algorithm also known as the Conjugated Structure – Algebraic Code Excited Linear Prediction (CS-ACELP) operating at a fixed rate of 8 kbps [8].

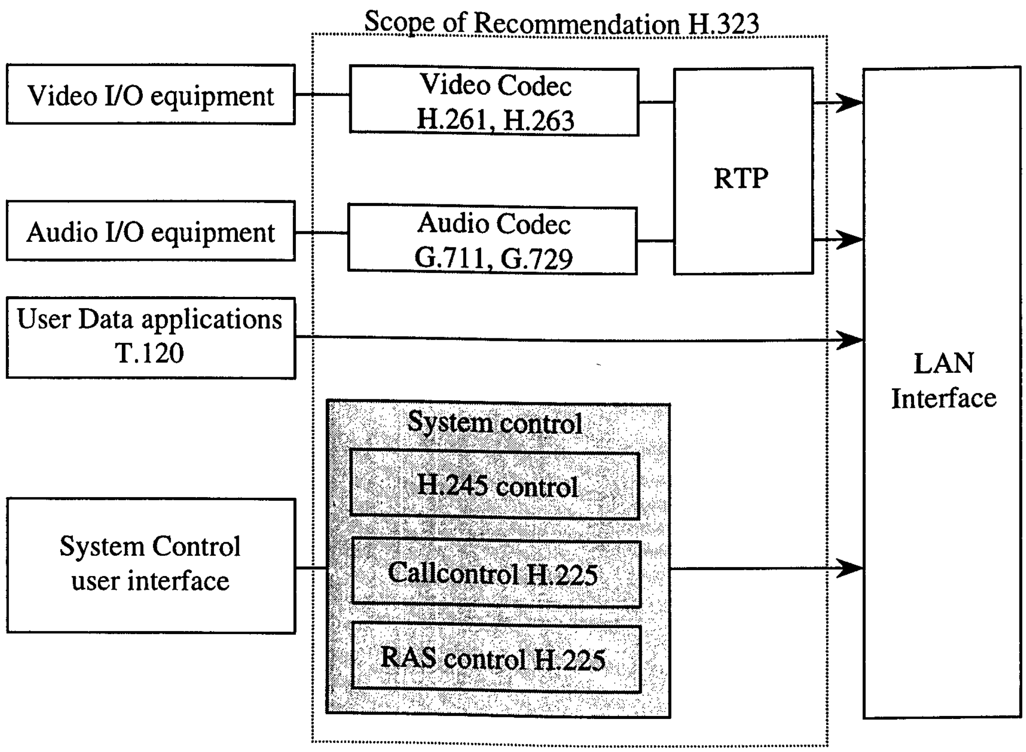

Figure 2 - Architectural Overview H.323 terminal equipment

Figure 3 shows the protocol stack for H.323/H.225 logical channels and other signalling. The transport, network, link and physical layers are a function of the LAN and are outside the scope of H.323. The protocols are shown in parenthesis and the scope of this report is shaded.

<table>
<tr><td>Audio apps</td><td>Video apps</td><td colspan="4">Terminal control and management</td><td>Data apps</td></tr>
<tr><td>G.711
G.722
G.728
G.729</td><td>H.261
H.263</td><td rowspan="2">RTCP</td><td rowspan="2">H.225
RAS channel</td><td>H.225
Call signalling channel</td><td>H.245
Control channel</td><td>T.124</td></tr>
<tr><td colspan="2">RTP</td><td colspan="2">X.224 Class 0</td><td>T.125</td></tr>
<tr><td colspan="4">Unreliable transport (UDP)</td><td colspan="2">Reliable transport (TCP)</td><td rowspan="3">T.123</td></tr>
<tr><td colspan="6">Network layer (IP)</td></tr>
<tr><td colspan="6">Link layer (PPP)</td></tr>
<tr><td colspan="7">Physical layer</td></tr>
</table>

Figure 3 - H.323 protocol stack

3.2.2 IP/UDP/RTP Header compression

Since RTP was published as an IETF Request for Comment (RFC) [9], there has been a growing interest in using RTP as one step towards achieving interoperability among different implementations of network audio/video applications. However, there is also concern that the 12-byte RTP header is too large an overhead for typical 20-byte payloads when operating over low speed links. Header size may be reduced through compression techniques as has been done with great success for TCP [10]. RTP header compression proposed in RFC 2508 [11] reduces the IP/UDP/RTP header in an RTP data packet from 40 bytes to approximately 2 to 4 bytes most of the time as shown in Figure 4. This big gain comes from the observation that although several fields change in every packet, the difference from packet to packet is often constant and therefore the second order difference is zero. By maintaining both the uncompressed header and the first order differences in the session state shared between the compressor and decompressor, all that must be communicated is an indication that the second order difference must be zero. The compressed packet carries a small integer, called the session context identifier or CID to indicate in which session context that packet should be interpreted. The CID is effectively an identification tag for the voice connection. Note that because RTP compression removes the IP header, it is likely to be negotiated and employed only on the physical link between routers. The IP header would need to be reconstituted in order to permit routing of the IP packet. Alternatively, layer three packet forwarding based on the CID might be possible between co-operating routers, nevertheless, this has no impact on the wide area bandwidth calculations in this report.

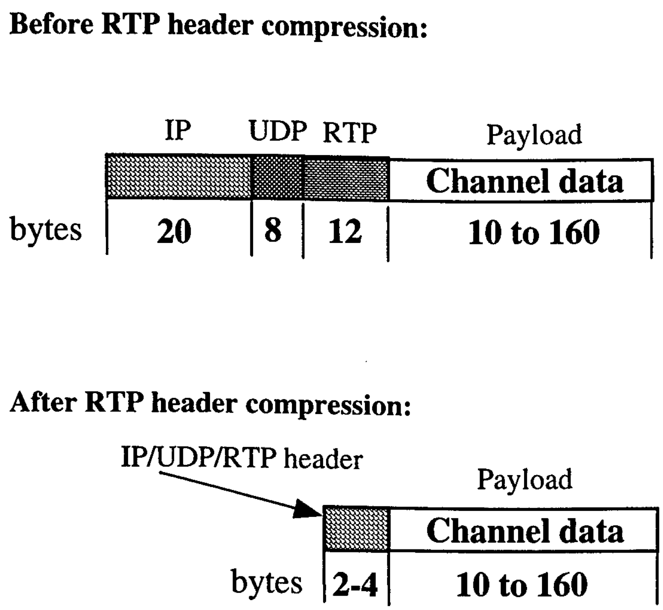

Figure 4 - IP/UDP/RTP header compression

3.2.3 Packet overhead and latency tradeoff

Increasing the number of frames per audio packet improves the bandwidth utilisation and decreases network packet overhead. However, it also introduces additional delay to the audio playback since a packet has to wait for all the audio frames to be accumulated before sending it across. Thus a tradeoff exists between packet overhead and local latency for audio packet transfer. Figure 5 displays the manner that efficiency, ie the percentage of audio data versus total number of bits in the RTP packet, increases as packet fill latency is permitted to increase.

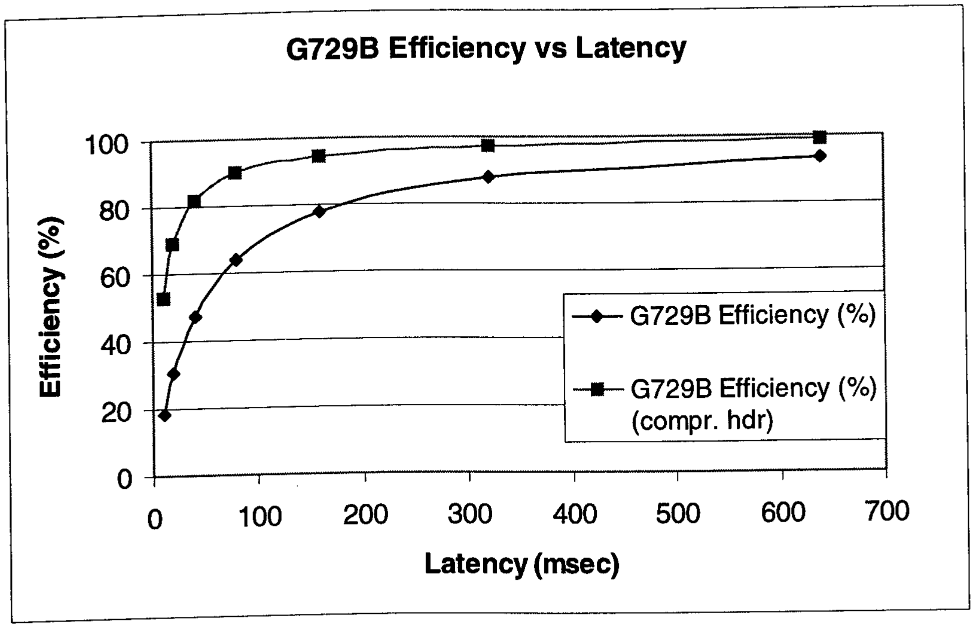

Figure 5 – G729B Efficiency (%) vs Latency

3.3 VTOA

Voice over ATM refers to the transport of voice and voice-band data over ATM. In this context "voice" refers to human speech, fax, and modem data. In order to provide a voice service over an ATM network, it is necessary to transport the voice plus associated signalling information.

3.3.1 The Issues

The simplest way to transport the voice traffic over the ATM would be to pass the entire inter-PABX trunk over an unstructured constant bit rate (CBR) virtual circuit using the ATM adaptation layer AAL-1. It is easy to implement this but it offers no scope for any bandwidth economy measures to reduce the impact of the unavoidable ATM overheads for instance by not transmitting idle channels. In order to achieve this flexibility in bandwidth allocation, an alternate approach would have each voice connection transferred over an individual ATM connection. While this would be quite feasible, there is one outstanding problem. With low rate voice coding schemes such as G.729B the time taken to fill an ATM cell leads to a substantial time latency (in the order of 50 ms).

3.3.2 AAL-2

As discussed before, we have conducted this examination using the AAL-2 standard intended for low bandwidth real-time variable bit rate (VBR) services. The AAL-2

standard is being developed in two fora, the International Telecommunications Union as ITU-T Recommendation I.363.2 [12] and I.366.2 [13], as well as the ATM Forum as AF-VTOA-0013.000 [14]. Figure 6 illustrates the hierarchy of standards cited in AF-VTOA-0013.000. SSCS refers to the service specific convergence sublayer that allows various services to be passed over the common part sublayer (CPS). For the actual voice data, the SSCS does not add any additional overheads. These are incurred only when transferring inband (DTMF tone dialling) signalling or fax signalling tones. The interworking facility to interworking facility (IWF-IWF) common channel signalling (CCS) can be transferred over either AAL-5 (ie treating the signalling as another data channel) or over the AAL-2 connection to which it is associated.

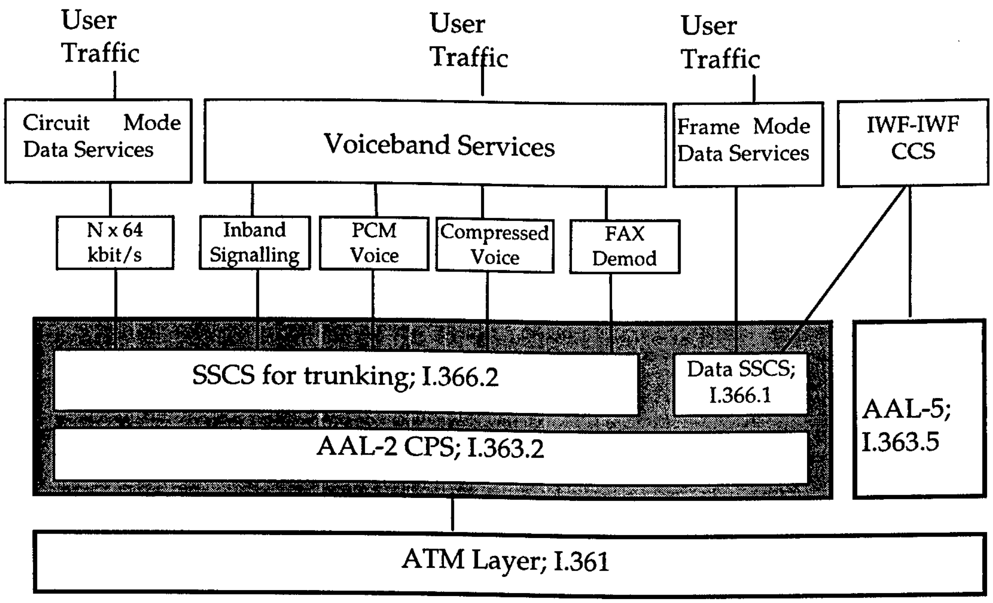

Figure 6 – VTOA (AAL-2) Standards

AAL-2 enables sub-multiplexing multiple voice channels, perhaps with each using a different voice coding standard, into a single ATM connection. This makes it ideal for use between voice switches (PABXs), where there are usually many simulataneous voice calls available to be aggregated. To discriminate between each of the calls, voice frames are pre-pended with a small (three octet) header (the AAL-2 CPS overhead). To avoid the excessive latency problem, AAL-2 employs two mechanisms. First, since all the PABX interconnection channels share the one ATM connection, ATM cells are filled at the aggregate rate of all the active channels. Second, if there are only a few channels active and cells are being delayed waiting for additional voice frames to complete the cell fill, the AAL-2 process can dispatch the (partially filled) cell at the expiry of a timer rather than exceed an acceptable latency limit. While this does waste some bandwidth, it only happens in extremely light traffic conditions. Figure 7 shows a typical voice packet structure segmented into ATM cells ready for transmission over the ATM path.

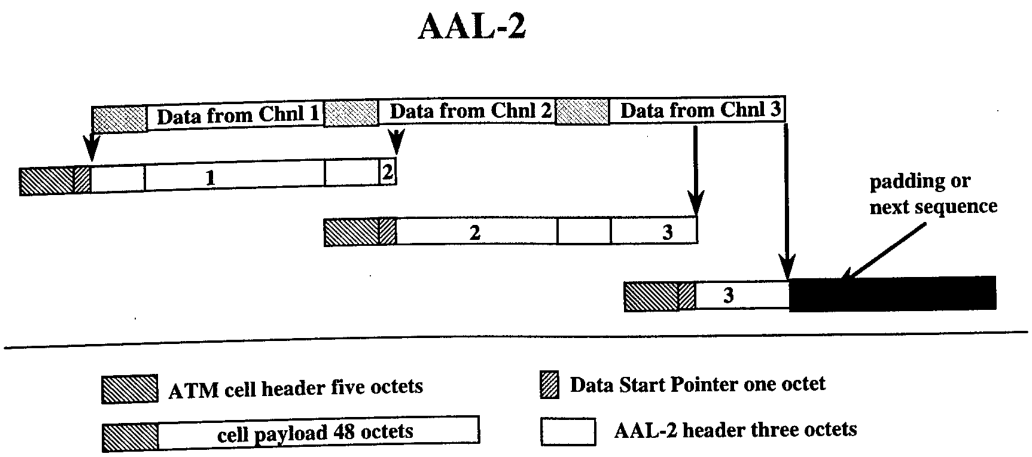

Figure 7 – AAL 2 over ATM

4. VoIP and VTOA Bandwidth Requirements

Annex A examines the demands of the two protocols. By comparison, the figure for the current Parakeet system is included. This TDM system provides 30 voice channels for 512 kbps consumed. It is important to note that the current Parakeet system employs a less efficient voice coding standard called continuously variable slope delta modulation (CVSD) which uses 16 kbps per channel rather than the 8 kbps of G.729B. Also, signalling traffic is not included in VoIP and VTOA analysis whereas the Parakeet TDM stream includes one 16 kbps channel carrying common channel signalling.

Results are summarised in Figure 8 which compares the VoIP and VTOA options with the Parakeet Eurocom TDM standard (30 channels totalling 512 kbps). AAL-2 is clearly the most economical compared with RTP with header compression or standard RTP with four voice frames per payload. The VoIP solution requires both header compression and multiple voice frames per payload to be comparable or better than VTOA.

Note that four voice frames per RTP payload was used for the analysis of multiple frame per payload as it appears to be the common choice within industry implementations. This is probably a compromise between efficiency and latency driven by the demands of maximum acceptable consecutive voice frame loss specification for G.729B. Calculations show that RTP with header compression and two voice frames per RTP payload is almost exactly equal to AAL-2.

The bandwidth demands analysis potentially gives a slightly optimistic figure for AAL2 for three or fewer channels active as it does not factor in the transmission of partially empty cells. However, the figures are accurate if it is acceptable to have up to four voice frame latency during low voice traffic periods.

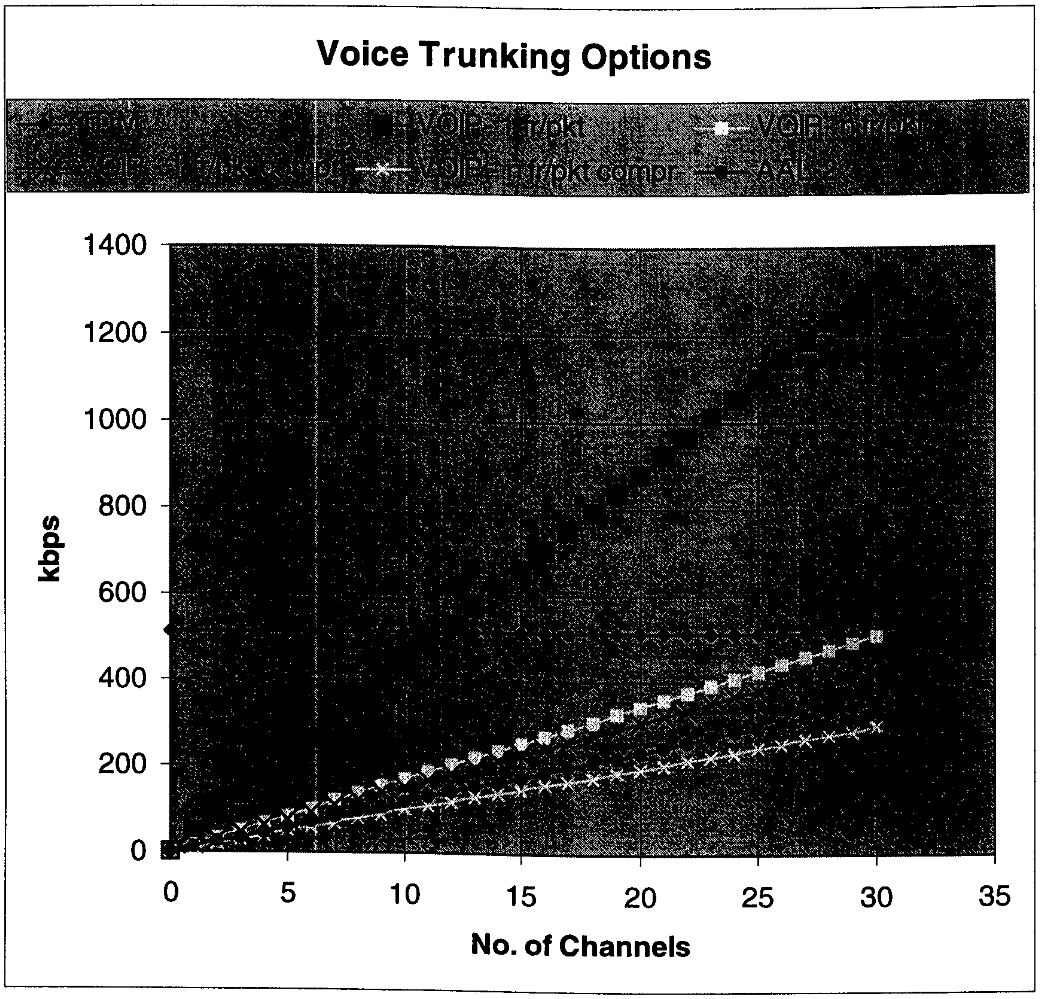

Figure 8 – VoIP VTOA protocol performance comparison, n=4

5. Signalling Requirements

The demands that a voice telephony system places on a network, specifically from the voice traffic it is carrying, were examined in Section 4. The other call upon network capacity comes from the need for signalling. The signalling system orchestrates the actual connection of users as well as housekeeping tasks, for instance the distribution of routing information. Again, this report has interest in the bandwidth requirements the signalling system has on the WAN. However, as is described in Annex B, the standards in this arena are, in parts, still being developed. Within the areas that are defined, the traffic demands are significantly less than those of the voice channel or are

unknown, determined by the network scenario. Accordingly, the signalling load will not be considered in this report.

6. Parakeet Reference Network

This section describes a reference network for Parakeet that is used in Appendix C to determine voice traffic load using the two candidate technologies. The network carries applications such as voice and general computer communications in a fully cryptographically secure manner. Circuit switching technology is central to the Parakeet system and all services pass through the circuit switch. The circuit switches are interconnected with TDM trunks. Each trunk circuit passes through an on-line government approved encryption device to provide communications security. Trunks typically run at 512 kbps and the rate cannot be changed without disruption to ongoing calls.

The reference network is shown in Figure 9 that has been extracted from the Parakeet specifications [15]. For the purposes of the specification, each node is given a unique identity. The 'S' prefix denotes a small circuit switch whilst 'L' denotes a large circuit switch – the size of the switch is largely irrelevant to this analysis. While the current phase of Parakeet will field only six ATM/IP equipped nodes, our analysis assumes all nodes are equipped with either ATM switches or IP routers.

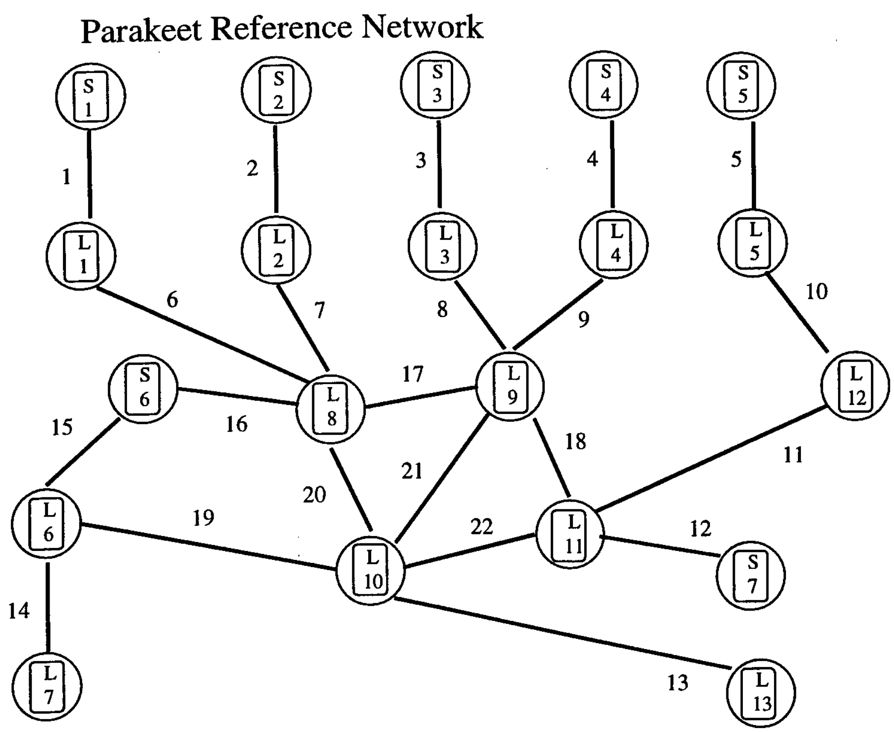

Figure 9 – Parakeet Reference Network

6.1 Traffic Load Calculation Procedures

Traffic load statistics were obtained from the Parakeet specifications [15]. Once traffic demands between the nodes have been allocated to pass over various trunks, the load on each trunk can be determined. In the operational network the routing decision will be via a flood search routing algorithm, typically this results in a shortest path selection. The busy hour call arrival rate for the busiest trunk link was then used in a conventional statistical model to determine the probability density function for the number of active channels in the busy hour [16].

6.2 Results

The detailed results including the traffic load calculation can be found in Appendix C. Table 1 shows the number of calls initiated in the busy hour for each trunk.

Link Name	Interconnect nodes	Number of calls initiated in the busy hour
1	S1/L1	107.5
2	S2/L2	130.0
3	S3/L3	125.0
4	S4/L4	125.0
5	S5/L5	107.5
6	L1/L8	169.5
7	L2/L8	208.3
8	L3/L9	207.0
9	L4/L9	207.0
10	L5/L12	167.0
11	L12/L11	239.5
12	L11/S7	37.5
13	L10/L13	85.0
14	L6/L7	96.3
15	L6/S6	70.4
16	L8/S6	167.7
17	L8/L9	239.2
18	L9/L11	122.9
19	L6/L10	156.0
20	L8/L10	156.3
21	L9/L10	111.5
22	L10/L11	137.8

Table 1 – Network Traffic Loading

Figure 10 shows the probability density function of the call activity on the busiest trunk. This shows that at no time does the demand on the busiest trunk exceed 25 channels of the 30 possible. The average number of channels active is between 11 and 12.

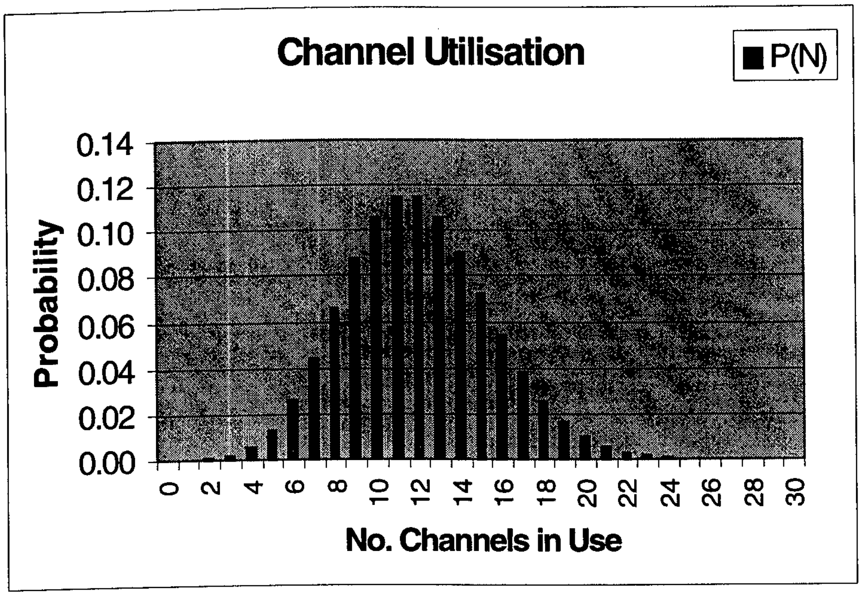

Figure 10 - Probability density function for busy hour call activity on the busiest trunk

7. Parakeet Voice via VoIP and VTOA

The VoIP and VTOA demands on Parakeet transmission capacity has been calculated in Appendix D. Peak demand (based on the peak of 25 active channels) on the busiest trunk is given in Table 2. For the system to be able to operate without undue packet latency or packet loss, the peak demands of the protocols must not exceed the total link rate. The expensive nature of standard VoIP over RTP is particularly telling.

TDM	VOIP 1fr/pkt	VOIP 4 fr/pkt	VOIP 1 fr/pkt compressed	VOIP 4 fr/pkt compressed	AAL-2
512	1100.0	425.0	380.0	245.0	293.2

Table 2 – Peak Bandwidth Demands (kbps)

To calculate an average bandwidth, the bandwidth demands of each protocol is weighted by the probability density function of the number of active channels. This provides a figure that indicates the average bandwidth being used by the voice system during the busy hour on the busiest trunk. The difference between this figure and the

total link rate indicates how much capacity would be available for non-real time communications. Table 3 shows the results for the options under examination.

TDM	VOIP 1fr/pkt	VOIP 4 fr/pkt	VOIP 1 fr/pkt compressed	VOIP 4 fr/pkt compressed	AAL-2
512	526.9	203.6	182.0	117.4	152.2

Table 3 – Average Bandwidth Demands (kbps)

8. Discussion and Conclusions

In this section we highlight some of the key contributions made in this report. The analysis has focussed on the bandwidth demands of the different approaches. Further work will be required to analyse the other factors (for instance network delays) that will also contribute to determining the feasibility of these approaches in the tactical domain.

8.1 RTP/UDP/IP Header Compression

Protocol header compression has been an active research area for the past couple of years especially after the maturity of the protocols and standards that drive audio and video streaming over the Internet. Any use of IP as a total WAN solution for voice services must implement RTP, UDP and IP header compression to be feasible in the tactical domain.

8.2 Bandwidth Savings

The average bandwidth demands used by the voice system during the busy hour on the busiest trunk indicate that VTOA (AAL-2) and VoIP with packet header compression are certainly promising solutions. If header compression is available and the use of four frames per packet (40 ms) does not lead to link delays unacceptable to users, VoIP can provide a slightly more economical solution than VTOA. Regardless of which approach is adopted, significant capacity can be released for non-real time communications.

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Appendix A: Network Load from Voice

ATM parameters in bits
24 AAL-2 frame header
80 AAL-2 frame size
0.28 equivalent ATM cells
10 frame latency (ms)

IP parameters in bits
40 PPP header - 5 bytes
160 IP header - 20 bytes
64 UDP header - 8 bytes
96 RTP header - 12 bytes
8 G.729 codec bit rate - kbits/sec
10 G.729 frame period - msec
80 G.729 data size per frame - bytes
4 Number of frames per packet
32 IP/UDP/RTP compressed header - 4 bytes

Voice over ATM
Because of the offset pointer, the number of octets per cell is 47.
The overall bit rate for voice over ATM using AAL-2 and a G729 speech coder (8kbps/sec) is given by,
(53/47)*Number_of_channels*((8/G279_frame_size)*(AAL2_header+ G729_frame_size))

Voice over IP
The overall bit rate for voice over IP is given by,

1 frame/packet:
Number_of_channels*(PPP_header+IP_header+UDP_header+RTP_header+G729_frame_size)/G729_frame_period

n frames/packet:
Number_of_channels*(PPP_header+IP_header+UDP_header+RTP_header+(n_frames_per_pkt*G729_frame_size))/(G729_frame_period*n_frames_per_pkt)

1 frame/packet (RTP/UDP/IP header compression):
Number_of_channels*(PPP_header+IP_UDP_RTP_compressed_header+G729_frame_size)/(G729_frame_period)

n frames/packet (RTP/UDP/IP header compression):
Number_of_channels*(PPP_header+IP_UDP_RTP_compressed_header+(n_frames_per_pkt*G729_frame_size))/(G729_frame_period*n_frames_per_pkt)

Channels active (N)	Bandwidth demands for given number of active channels (N)					
	TDM	VOIP- 1 fr/pkt	VOIP- 4 fr/pkt	VOIP - 1 fr/pkt compr	VOIP - 4 fr/pkt compr	AAL-2
0	512	0.0	0.0	0.0	0.0	0.0
1	512	44.0	17.0	15.2	9.8	11.7
2	512	88.0	34.0	30.4	19.6	23.5
3	512	132.0	51.0	45.6	29.4	35.2
4	512	176.0	68.0	60.8	39.2	46.9
5	512	220.0	85.0	76.0	49.0	58.6
6	512	264.0	102.0	91.2	58.8	70.4
7	512	308.0	119.0	106.4	68.6	82.1
8	512	352.0	136.0	121.6	78.4	93.8
9	512	396.0	153.0	136.8	88.2	105.5
10	512	440.0	170.0	152.0	98.0	117.3
11	512	484.0	187.0	167.2	107.8	129.0
12	512	528.0	204.0	182.4	117.6	140.7
13	512	572.0	221.0	197.6	127.4	152.5
14	512	616.0	238.0	212.8	137.2	164.2
15	512	660.0	255.0	228.0	147.0	175.9
16	512	704.0	272.0	243.2	156.8	187.6
17	512	748.0	289.0	258.4	166.6	199.4
18	512	792.0	306.0	273.6	176.4	211.1
19	512	836.0	323.0	288.8	186.2	222.8
20	512	880.0	340.0	304.0	196.0	234.6
21	512	924.0	357.0	319.2	205.8	246.3
22	512	968.0	374.0	334.4	215.6	258.0
23	512	1012.0	391.0	349.6	225.4	269.7
24	512	1056.0	408.0	364.8	235.2	281.5
25	512	1100.0	425.0	380.0	245.0	293.2
26	512	1144.0	442.0	395.2	254.8	304.9
27	512	1188.0	459.0	410.4	264.6	316.6
28	512	1232.0	476.0	425.6	274.4	328.4
29	512	1276.0	493.0	440.8	284.2	340.1
30	512	1320.0	510.0	456.0	294.0	351.8

Appendix B: Signalling Requirement

B.1 VOIP

H.323 defines two basic models for call signalling. The first model is direct call signalling, where signalling takes place between the endpoints (terminals). The second model is gatekeeper routed call signalling, where the gatekeeper relays all call signalling between endpoints. Call signalling procedures are based on Q.931 as tailored by H.225.0. There are three signalling scenarios that are derived from the basic models:

- endpoint 1 - endpoint 2 communications (direct model).
- endpoint 1 - gatekeeper - endpoint 2 communications (gatekeeper mediated). Negotiating a connection through a common gatekeeper would be typically only appropriate when both endpoints were on the same LAN and hence will not be considered further in this report.
- endpoint 1 - gatekeeper 1 - gatekeeper 2 - endpoint 2 communications (gatekeeper mediated). Such scenario would see a gatekeeper on each LAN acting as the controlling function usually seen in the PABX of traditional voice networks.

Figure 11 shows the steps involved in establishing a call for scenario 1.

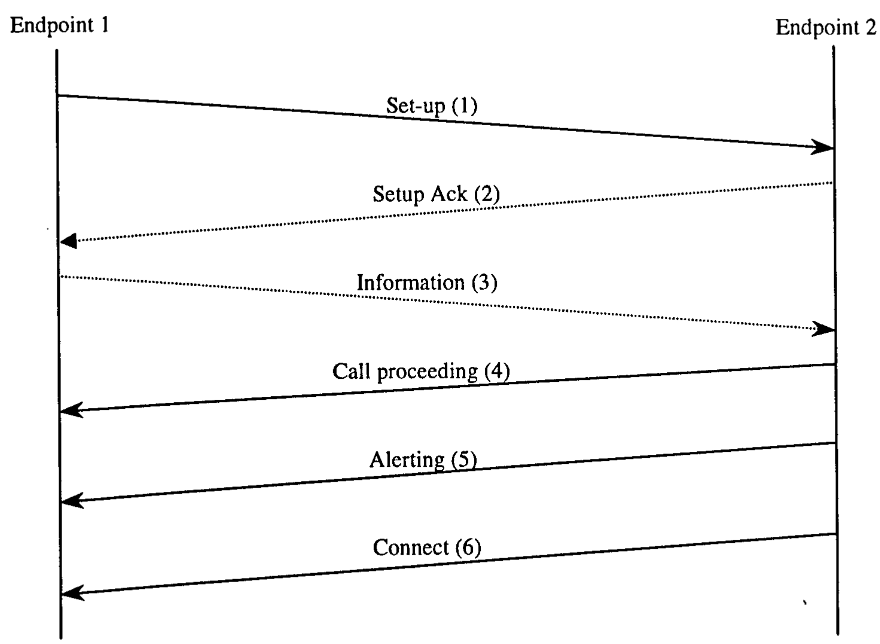


Figure 11 - Call signalling scenario 1 – endpoint1-endpoint2 (direct)

Table 4 illustrates the size of typical success, failure and status messages (in bytes) traversing the WAN. For example, in a successful call setup, the maximum amount of

signalling information is about 691 bytes. Note that the maximum values were obtained when the mandatory fields were completely utilised A typical message structure is shown in Figure 12. For a nominal 3 minute call duration this signalling load is minimal (averaging 30 bps).

Message Size (bytes)	Success	Failure	Status
Minimum	45	43	17
Maximum	691	558	45

Table 4 - Message Size in bytes for success, failure and status messages

Setup Message

Information element	status	Length
Protocol discriminator	M	1
Call reference	M	3
Message type	M	1
Sending complete	O	1
Bearer capability	M	5-6
Display	O	2-82
Keypad facility	O	2-34
Signal	O	2-3
Calling party number	O	2-131
Called party number	O	2-131
User-to-User	M	2-131

Setup-UUIE :

- Protocol Identifier
- H245 Address
- Source Address
- Source Info
- Dest Address
- Dest Call Signal Address
- Active MC
- Conference ID
- Conference Goal
- Call Services
- Call Type
- ...

Figure 12 - Setup messsage structure in bytes, M for mandatory, O for optional

The final scenario is of particular interest to us. To date gatekeeper to gatekeeper communications have not yet been standardised. The draft standard, [17] discusses a framework for a peer gatekeeper routing protocol (PGRP) that supports the exchange of information among gatekeepers about elements in their respective zones (see Figure 13). PGRP provides a mechanism by which gatekeepers may acquire knowledge of both static and dynamic information from other zones. At each level of the hierarchy, information is collated and a summary of this information is then distributed.

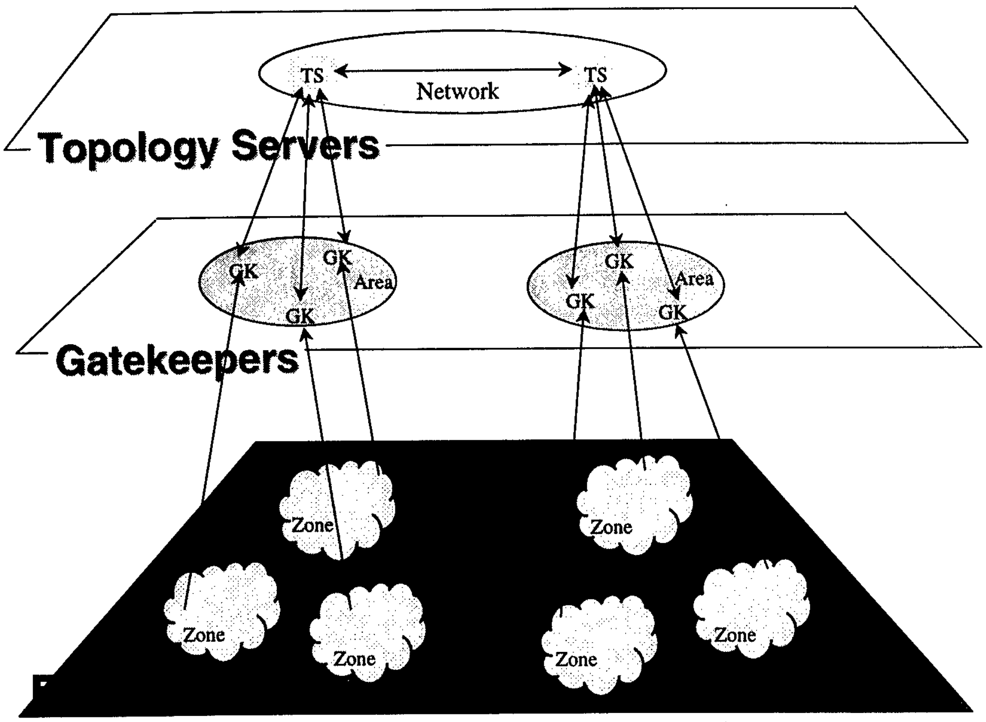


Figure 13 - Scenario 3 - endpoint – gatekeeper – gatekeeper - endpoint communications

After call setup is complete, all communications between endpoints take place over logical channels that are opened using the procedures of H.245. There is only one logical channel for control and separate logical channels for each media type. In addition to the logical channels opened for the actual audio or video media, a separate logical channel for the RTCP must be opened to provide a feedback mechanism for QoS to the source media.

B.2 VTOA

Signalling standards between voice circuit switches are covered by a number of techniques based on ITU standard Q.931 including Signalling System 7, Digital Private Network Signalling System (DPNSS) and the ETSI QSIG system. Note that QSIG is now being standardised as Private Signalling System no. 1 (PSS1) by ISO/IEC in ISO/IEC 11572. These systems typically employ one of the voice channel time slots to provide the signalling on a common channel between the two switches. If a network of voice switches is simply interconnected over an ATM network, then it is relatively simple to transfer the common channel signalling as a constant bit rate 64 kbps service.

For this report, the preferred configuration sees the ATM network acting as a virtual voice switch. In this case, the common channel signalling is terminated and interpreted in the IWF. The signalling that passes between the IWF and the local voice switch comprises two fundamental elements: basic services such as call establishment/tear down and supplementary services such as call redirection. The

ATM Forum [14] specifies that IWFs should be able to interchange PSS1 messages via either AAL-5 or AAL-2 data channels. This report has not attempted to examine the traffic load generated by PSS1 over ATM. Apart from the call setup/tear down which might be analysed, there are a number of scenario dependent elements which would be difficult to itemise such as signalling to establish switched virtual circuits between ATM elements versus provision via permanent virtual circuits, sharing of routing information etc. What is standardised in AAL-2 [12] is the interchange to establish/remove voice connections in the AAL-2 structure. These are passed using the AAL-2 protocol itself using a logical channel associated with each voice channel. The data elements are very small. For instance the channel assignment request message has mandatory elements totalling only 3 octets.

Appendix C: Voice Traffic Model

Traffic Statistics provided by the Parakeet Specification (per day call initiation)

Src\Dest	S1	S2	S3	S4	S5	S6	S7	L1	L2	L3	L4	L5	L6	L7	L8	L9	L10	L11	L12	L13	Tot
S1	165	20				40		40	10				10	5	30	20	30				370
S2	20	165	20			40		20	40	20			10	5	30	20	30				420
S3		20	165	20		40			20	40	20		10	5	30	20	30				420
S4			20	165	20	40				20	40	20	10	5	30	20	30				420
S5				20	165	40					20	40	10	5	30	20	30				380
S6	40	40	40	40	40	185	35	10	10	10	10	10	30	35							535
S7						35	165						10	5				35			250
L1	30	20				10		95	35				10	10	10	20	20		15	13	288
L2	10	30				10		35	95	35			10	5	10	20	20		15	13	308
L3		10	30			10			35	95	35		10	10	10	20	20		15	13	313
L4			10	30		10				35	95	35	10	10	10	20	20		15	13	313
L5				10	30	10					35	95	10	10	10	20	20		15	13	278
L6	10	10	10	10	10	30		10	10	10	10	10	185	35	15	15	15		35	25	455
L7	10	10	10	10	10	30		10	10	10	10	10	30	160					30	20	370
L8	25	25	25	25	25	10		20	20	20	20	20	15	10	150	40	40		15	15	520
L9	25	25	25	25	25	25		20	20	20	20	20	25		40	150	40		15	15	535
L10	25	25	25	25	25	25		20	20	20	20	20	25		40	40	150		15	15	535
L11							30											165			195
L12	15	15	15	15	15	15		5	5	5	5	5		10	10	10			160	15	320
L13	15	15	15	15	15	15		5	5	5	5	5		10	10	10	10		15	160	330
																					0
Totals	390	430	410	410	380	620	230	290	335	345	345	290	420	335	465	465	505	200	360	330	

The Parakeet Specification calls for the Busy Hour to have four times the average hourly number of call initiations. The source to destination determines (via shortest

path) the links that each call initiation is to traverse. One can then calculate the number of calls processed in the busy hour on the busiest link.

Link Utilisation

Link Name	Interconnect nodes	Number of calls initiated in the busy hour
1	S1/L1	107.5
2	S2/L2	130.0
3	S3/L3	125.0
4	S4/L4	125.0
5	S5/L5	107.5
6	L1/L8	169.5
7	L2/L8	208.3
8	L3/L9	207.0
9	L4/L9	207.0
10	L5/L12	167.0
11	L12/L11	239.5
12	L11/S7	37.5
13	L10/L13	85.0
14	L6/L7	96.3
15	L6/S6	70.4
16	L8/S6	167.7
17	L8/L9	239.2
18	L9/L11	122.9
19	L6/L10	156.0
20	L8/L10	156.3
21	L9/L10	111.5
22	L10/L11	137.8

The following procedures are followed to determine the channel utilisation

30 - number of channels (K)
239.5 - mean arrival rate per hour
3.991667 - mean arrival rate per minute, λ
3 - mean call holding time in minutes
0.333333 - mean service rate per minute, μ
11.975 - utilization factor for each channel, ρ
0.399167 - utilization factor for system, $\overline{\rho}$
6.3E-06 - probability of no calls, *P(0)*

The following formulas were used to compute the measures of performance [16]:

Utilisation factor of the entire system, $\overline{\rho}$:

$$\overline{\rho} = \frac{\rho}{K} = \frac{\lambda}{K\mu}$$

Assuming that $\lambda<K\mu$ (a necessary condition to avoid an explosive queue),
The probability of no calls, *P(0)*:

$$P(0) = \frac{1}{\dfrac{\rho^K}{K!(1-\overline{\rho})} + \sum_{i=0}^{K-1} \dfrac{\rho^i}{i!}}$$

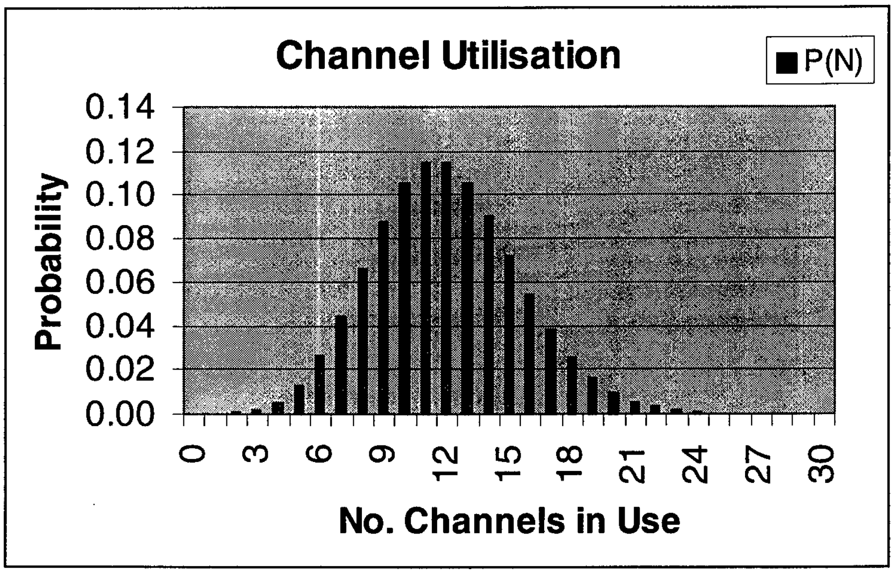
The probability of N channels being in use, *P(N)*:

$$P(N) = P(0)\frac{\rho^N}{N!}, \text{if } N \le K$$

$$P(N) = \frac{P(0)\overline{\rho}K^K}{K!}, \text{if } N > K$$

Channel Utilisation

Channels, N	P(N)	Cum Dist
0	0.000	0.000
1	0.000	0.000
2	0.000	0.001
3	0.002	0.002
4	0.005	0.008
5	0.013	0.021
6	0.026	0.046
7	0.044	0.091
8	0.066	0.157
9	0.088	0.245
10	0.105	0.350
11	0.115	0.464
12	0.114	0.579
13	0.105	0.684
14	0.090	0.774
15	0.072	0.846
16	0.054	0.900
17	0.038	0.938
18	0.025	0.963
19	0.016	0.979
20	0.010	0.989
21	0.005	0.994
22	0.003	0.997
23	0.002	0.999
24	0.001	0.999
25	0.000	1.000
26	0.000	1.000
27	0.000	1.000
28	0.000	1.000
29	0.000	1.000
30	0.000	1.000


Channel Utilisation
P(N)
Probability
0.14
0.12
0.10
0.08
0.06
0.04
0.02
0.00
0
3
6
9
12
15
18
21
24
27
30
No. Channels in Use

Appendix D: VoIP and VTOA Demands on Parakeet Transmission

Table applies the traffic analysis to the bandwidth demands.

Chn actv (N)	Bandwidth demands for given number of active channels (N)						P(N)	Contribution to Average Bandwidth demand				
	TDM	VOIP- 1 fr/pkt	VOIP- 4 fr/pkt	VOIP - 1 fr/pkt compr	VOIP - 4 fr/pkt compr	AAL-2		VOIP- 1 fr/pkt	VOIP- 4 fr/pkt	VOIP - 1 fr/pkt compr	VOIP - 4 fr/pkt compr	AAL-2
0	512	0.0	0.0	0.0	0.0	0.0	0.000	0.0	0.0	0.0	0.0	0.0
1	512	44.0	17.0	15.2	9.8	11.7	0.000	0.0	0.0	0.0	0.0	0.0
2	512	88.0	34.0	30.4	19.6	23.5	0.000	0.0	0.0	0.0	0.0	0.0
3	512	132.0	51.0	45.6	29.4	35.2	0.002	0.2	0.1	0.1	0.1	0.1
4	512	176.0	68.0	60.8	39.2	46.9	0.005	1.0	0.4	0.3	0.2	0.3
5	512	220.0	85.0	76.0	49.0	58.6	0.013	2.8	1.1	1.0	0.6	0.8
6	512	264.0	102.0	91.2	58.8	70.4	0.026	6.8	2.6	2.4	1.5	1.8
7	512	308.0	119.0	106.4	68.6	82.1	0.044	13.6	5.3	4.7	3.0	3.6
8	512	352.0	136.0	121.6	78.4	93.8	0.066	23.3	9.0	8.0	5.2	6.2
9	512	396.0	153.0	136.8	88.2	105.5	0.088	34.8	13.5	12.0	7.8	9.3
10	512	440.0	170.0	152.0	98.0	117.3	0.105	46.3	17.9	16.0	10.3	12.3
11	512	484.0	187.0	167.2	107.8	129.0	0.115	55.5	21.4	19.2	12.4	14.8
12	512	528.0	204.0	182.4	117.6	140.7	0.114	60.4	23.3	20.9	13.4	16.1
13	512	572.0	221.0	197.6	127.4	152.5	0.105	60.3	23.3	20.8	13.4	16.1
14	512	616.0	238.0	212.8	137.2	164.2	0.090	55.5	21.4	19.2	12.4	14.8
15	512	660.0	255.0	228.0	147.0	175.9	0.072	47.5	18.3	16.4	10.6	12.7
16	512	704.0	272.0	243.2	156.8	187.6	0.054	37.9	14.6	13.1	8.4	10.1
17	512	748.0	289.0	258.4	166.6	199.4	0.038	28.4	11.0	9.8	6.3	7.6
18	512	792.0	306.0	273.6	176.4	211.1	0.025	20.0	7.7	6.9	4.5	5.3
19	512	836.0	323.0	288.8	186.2	222.8	0.016	13.3	5.1	4.6	3.0	3.5
20	512	880.0	340.0	304.0	196.0	234.6	0.010	8.4	3.2	2.9	1.9	2.2
21	512	924.0	357.0	319.2	205.8	246.3	0.005	5.0	1.9	1.7	1.1	1.3
22	512	968.0	374.0	334.4	215.6	258.0	0.003	2.9	1.1	1.0	0.6	0.8
23	512	1012.0	391.0	349.6	225.4	269.7	0.002	1.6	0.6	0.5	0.3	0.4
24	512	1056.0	408.0	364.8	235.2	281.5	0.001	0.8	0.3	0.3	0.2	0.2
25	512	1100.0	425.0	380.0	245.0	293.2	0.000	0.4	0.2	0.1	0.1	0.1
26	512	1144.0	442.0	395.2	254.8	304.9	0.000	0.2	0.1	0.1	0.0	0.1
27	512	1188.0	459.0	410.4	264.6	316.6	0.000	0.1	0.0	0.0	0.0	0.0
28	512	1232.0	476.0	425.6	274.4	328.4	0.000	0.0	0.0	0.0	0.0	0.0
29	512	1276.0	493.0	440.8	284.2	340.1	0.000	0.0	0.0	0.0	0.0	0.0
30	512	1320.0	510.0	456.0	294.0	351.8	0.000	0.0	0.0	0.0	0.0	0.0
Ave	512							526.9	203.6	182.0	117.4	152.2

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19. ABSTRACT

The current generation Battlespace Communications System (Land) for voice (telephony) is based on a military standard circuit switching technology. There has been considerable discussion on alternatives based on packetised voice, ie voice over Internet Protocol and voice over ATM. This paper examines these two options and considers the bandwidth demands of each option, specifically in the context of an agreed reference model of the network including traffic load. This report is preliminary work prior to the combination of this voice model with that of an agreed data workload and examination through simulation of consequent integrated network performance.